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About This Guide

This guide provides instructions on how to configure an APX™ or MAX TNT® to process MultiVoice voice over IP (VoIP) calls.

Note This manual describes the full set of features for units running software version TAOS 10.0. Some features might not be available with earlier versions or specialty loads of the software.

The APX family of products includes multiple platforms that differ in call capacity and hardware, but support the same operating system and similar configuration options. The APX family, which includes the APX 8000 and APX 1000 products, shares many features with its MAX TNT predecessor. For features that are supported with no differences across all the platforms, this manual often refers to your product as a *TAOS unit.*

Warning Before installing your unit, be sure to read the safety instructions in the *Edge Access and Broadband Access Safety and Compliance Guide.* For information specific to your unit, see the "Safety-Related Electrical, Physical, and Environmental Information" appendix in your unit's hardware installation guide or *Getting Started Guide.*

What you need to know

This manual is intended for the person who configures and maintains your TAOS unit running MultiVoice. To use the manual effectively, you must have a basic understanding of security and configuration, and be familiar with authentication servers and networking concepts. You also need to understand Internet and telecommuting concepts and dial-in connections (both framed protocol sessions and user logins).

Following are the special characters and typographical conventions used in this manual:

Convention	Meaning
`Monospace text`	Represents text that appears on your computer's screen, or that might appear on your computer's screen.
`Boldface monospace text`	Represents characters that you enter exactly as shown (unless the characters are also in ***`italics`***—see *Italics*, following). If you can enter the characters but are not specifically instructed to, they do not appear in boldface.

Convention	Meaning
Italics	Represent variable information. Do not enter the words themselves in the command. Enter the information they represent. In ordinary text, italics are used for titles of publications, for some terms that would otherwise be in quotation marks, and to show emphasis.
[ ]	Indicate an optional argument you might add to a command. To include such an argument, type only the information inside the brackets. Do not type the brackets unless they appear in boldface.
|	Separates command choices that are mutually exclusive.
>	Separates levels of profiles, subprofiles, and parameters in a hierarchical menu when the path to a menu item is referred to in text.
:	Separates levels of profiles, subprofiles, and parameters in a pathname displayed in the command-line interface or referred to in text.
Key1+Key2	Represents a combination keystroke. To enter a combination keystroke, press the first key and hold it down while you press one or more other keys. Release all the keys at the same time. (For example, Ctrl+H means hold down the Ctrl key and press the H key.)
Press Enter	Means press the Enter or Return key or its equivalent on your computer.
Note:	Introduces important additional information.
Caution:	Warns that a failure to follow the recommended procedure can result in loss of data or damage to equipment.
Warning:	Warns that a failure to take appropriate safety precautions can result in physical injury.
Warning:	Warns of danger of electric shock.

Documentation set

The documentation set for APX and MAX TNT products consists of the following manuals, available at `http:www.lucent.com/support` and `http//:www.lucentdocs.com/ins`:

- **Read me first:**

   - *Edge Access and Broadband Access Safety and Compliance Guide.* Contains important safety instructions and country-specific compliance information that you must read before installing a unit.
   - *TAOS Command-Line Interface Guide.* Introduces the TAOS command-line environment and shows how to use the command-line interface effectively. This manual describes keyboard shortcuts and introduces commands, security levels, profile structure, and parameter types.
- **Installation and basic configuration:** *Getting Started Guide* or hardware installation guide for your unit. Shows how to install the unit's chassis and hardware, and includes technical specifications. A *Getting Started Guide* also shows you how to provide the basic configuration needed to access the unit on a network.
- **Configuration:**
   - *Physical Interface Configuration Guide* for your unit. Describes how to provision the slot cards supported in the unit, and how to configure the cards' physical interfaces. This guide also describes system allocation of slot card resources, and how to use the supported cards in a variety of data environments.
   - *APX/MAX TNT ATM Configuration Guide.* Describes how to configure Asynchronous Transfer Mode (ATM) permanent virtual circuit (PVC) and switched virtual circuit (SVC) ATM interfaces. It includes information about ATM direct and ATM-frame relay circuits.
   - *APX/MAX TNT Frame Relay Configuration Guide.* Describes how to configure frame relay operations on a unit. This guide explains physical layer restrictions and how to create permanent virtual circuit (PVC) and switched virtual circuit (SVC) interfaces. It includes information about Multilink frame relay (MFR) and link management, as well as frame relay and frame relay direct circuits.
   - *APX/MAX TNT WAN, Routing, and Tunneling Configuration Guide.* Shows how to configure LAN and WAN routing for analog and digital dial-in connections on a unit. This guide includes information about IP routing, Open Shortest Path First (OSPF) routing, Border Gateway Protocol (BGP) routing, Internet Group Management Protocol (IGMP) routing, multiprotocol routers, virtual routers (VRouters), and tunneling protocols.
- **MultiVoice:**
   - *MultiVoice® for APX/MAX TNT Configuration Guide.* Shows how to configure the MultiVoice® application to run on a unit in both Signaling System 7 (SS7) and H.323 Voice over IP (VoIP) configurations.
   - *MultiVoice Access Manager User's Guide.* Describes the installation, configuration, and administration of MultiVoice Access Manager, which provides H.323 gatekeeper functions for MultiVoice networks.
- **RADIUS:** *TAOS RADIUS Guide and Reference.* Describes how to set up a unit to use the Remote Authentication Dial-In User Service (RADIUS) server, and contains a complete reference to RADIUS attributes.
- **Administration and troubleshooting:** *APX/MAX TNT Administration Guide.* Describes how to administer a unit, including how to monitor the system and cards, troubleshoot the unit, and configure the unit to use the Simple Network Management Protocol (SNMP).

- **Reference:**
  - *APX/MAX TNT Reference*. An alphabetic reference to all commands, profiles, and parameters supported on a unit.
  - *TAOS Glossary*. Defines terms used in the documentation for a unit.

Related publications

This guide and documentation set do not provide a detailed explanation of products, architectures, or standards developed by other companies or organizations. Following are some publications that you might find useful:

- ITU Telecommunication sector standard (ITU-T) H.323, *Packet-based multimedia communications systems* (Feb. 1998), International Telecommunications Union.
- RFC 1889, *RTP: A Transport Protocol for Real-Time Applications* (Jan. 1996), IETF.
- RFC 2705, *Media Gateway Control Protocol (MGCP)* (Oct. 1999), IETF.
- *Signaling in Today's Telecommunication Networks*, John G. van Bosse.
- *Delivering Voice over IP Networks*, Dan Minoli, Emma Minoli, Daniel Minoli.
- *Delivering Voice Over Frame Relay and ATM*, Dan Minoli.
- *The Guide to T1 Networking*, William A. Flanagan.
- *TCP/IP Illustrated*, W. Richard Stevens.
- *Firewalls and Internet Security*, William R. Cheswick and Steven M. Bellovin.

Following are some related World Wide Web (WWW) sites:

- `http://www.ietf.org/rfc`
- `http://www.itu.ch/`
- `http://www.cs.columbia.edu/~hgs/rtp/drafts/VoIP97-8.pdf`
- `http://www.cs.columbia.edu/~hgs/rtp/`

Note The listed web sites were available at the time of this manual's publication. Lucent does not maintain these sites and cannot guarantee their availability in the future.

Introducing MultiVoice Concepts

1

The public switched telephone network 1-1

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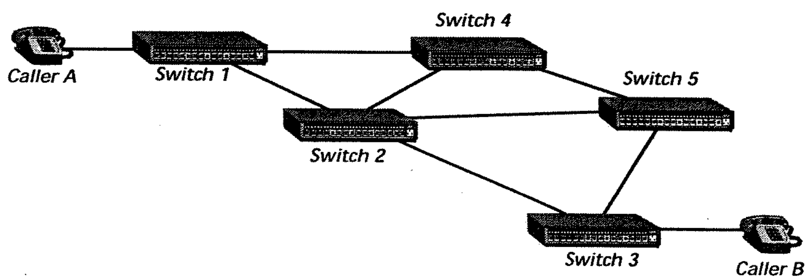
The public switched telephone network

Traditionally, real-time voice information is sent over the public switched telephone network (PSTN). Circuit-switched technology provides every call with dedicated bandwidth, usually 64Kbps. End-to-end calls are established on the basis of a sequence of dialed digits, and the PSTN dedicates a physical path between callers. Because the telephone equipment establishes the call path at the beginning of the call, the path can change *between* calls, but never while a call is active.

Figure 1-1 illustrates an example of a PSTN network. Caller A dials Caller B's phone number. As Caller A dials the phone number, the network might route the call from Switch 1 to Switch 2 to Switch 3, which connects to Caller B. Once the PSTN establishes the call, communication travels only through Switch 1, Switch 2, and Switch 3.

Figure 1-1. Example of call routing over circuit-switched PSTN



If Caller A dials Caller B again, the PSTN might establish the call by routing it from Switch 1 to Switch 4 to Switch 5 to Switch 3 before finally connecting Caller A to Caller B. Again, the path can change between calls, but not during any specific call.

In contrast, an Internet Protocol (IP) network has a packet-switched architecture. Devices transmit data in packets, and the path of one packet from end to end can vary from another packet within an established session. In addition to data, packets

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contain addressing information, which routing devices use to send each packet to its destination. Routing devices maintain tables that instruct them how to direct packets. Dynamic protocols, like Routing Information Protocol (RIP) or Open Shortest Path First (OSPF), define methods that routing devices use to update each other as networking environments change.

In the past, the PSTN was the only network supporting voice communication. With MultiVoice, voice traffic can be transmitted across IP networks.

The MultiVoice network

MultiVoice complies with International Telecommunications Union Telecommunication Standardization sector standard (ITU-T) H.323 for transmitting voice telephone calls across IP networks. The H.323 standard defines a framework for the transmission of real-time voice communications across IP networks. MultiVoice on the APX or MAX TNT also supports integration with Signaling System 7 (SS7) networks by means of IP Device Control (IPDC), a media gateway control protocol, to provide call control for Voice over IP calls originating from SS7 networks.

Multivoice terms and definitions

In addition to the vocabulary used in the TAOS environment, MultiVoice uses some specific voice-related expressions. The following lists the most common terms with their definitions.

Term	Definition
Call end-point	The communications device used to initiate or answer a call, or a call's origin or destination.
Egress	A general voice-related term for an exit. For MultiVoice, a location or device used to route data from the packet network onto the analog network. Related terms: egress PSTN, egress switch
Egress gateway	A term specific to MultiVoice for the TAOS unit which connects a VoIP call to the called telephone number. The egress gateway: • Dials the destination telephone number • Converts data from the packet network to analog voice • Reports call progress Related terms: egress MultiVoice Gateway, egress TAOS unit
Far end	A general voice-related term for the remote call termination point, relative to the active call end point. For MultiVoice, the location or device—relative to the point-of-origin of network packets, call signals, etc.—where packets, call signals, etc., for the remote call end point are processed. Related terms: far-end PSTN, far-end switch
Far-end gateway	(Specific to MultiVoice) The TAOS unit at the opposite end of the packet network connection—relative to the active call end point. Related terms: far-end MultiVoice Gateway, far-end TAOS unit

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Term	Definition
Ingress	(General) An entrance. For MultiVoice, a location or device where voice signals from the analog network are routed onto the packet network. Related terms: ingress PSTN, ingress switch
Ingress gateway	(Specific to MultiVoice) The TAOS unit where a VoIP call originates. The ingress gateway: • Accepts calls from the PSTN • Initiates requests for call admissions • Converts analog voice to packet network data • Reports call progress Related terms: ingress MultiVoice Gateway, ingress TAOS unit
MultiVoice Access Manager (MVAM)	A MultiVoice component that supports the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) H.323 standard for managing an IP network. MVAM supports MultiVoice Gateways, user profiles, and authentication. Capabilities supported by MVAM include phone-to-IP address translation, Web-based administration interface, PIN-based user authentication, virtual private network (VPN) support, Telephone number aliases, call detail reporting (CDR), Gateway and user database support, and third-party billing system support.
MultiVoice Gateway	A MultiVoice component that supports the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) H.323 standard for transmitting voice over an IP network. When a voice call is received at a local MultiVoice Gateway, the voice signal is packetized, compressed, and transmitted over the packet network using standard protocols and voice-compression technologies. At the remote gateway, the process is reversed and the call is delivered over the remote packet network to its intended destination.
Near end	(General) A local call termination point, relative to the active call end point. For MultiVoice, the location or device—relative to the point-of-origin of network packets, call signals, etc.—where packet processing, call signaling, etc., is initiated for the active call end point. Related terms: near-end PSTN, near-end switch
Near-end gateway	(Specific to MultiVoice) The TAOS unit which provides the local connection to the packet network—relative to the active call end point. Related terms: near-end MultiVoice Gateway, near-end TAOS unit

MultiVoice packet processing

MultiVoice voice and data are processed using User Datagram Protocol (UDP) packets. UDP is a protocol within the TCP/IP protocol suite that is used in place of TCP for processing real-time audio and video traffic. UDP is used with Real-time Transport Protocol (RTP) to provide delivery, packet sequence checking, and error notification for MultiVoice call processing.

Voice over IP (VoIP) call data is compressed into frames assembled inside RTP packets. Each RTP packet is wrapped within a UDP packet and includes timestamping and synchronization information in its header for proper reassembly of the voice frames at the receiving end.

In a UDP/IP stack, the RTP header is created first and then the packet is moved down the stack to UDP and IP. UDP hands over these packets to the IP protocol layer along with the IP address of the destination node.

At the IP layer, the target address information for the destination gateway is processed, then passed to the Ethernet layer which establishes the data link between the two MultiVoice Gateways; completing the connection between the packet network and the PSTN.

Figure 1-2 illustrates how each MultiVoice packet is formatted for transmission across the packet network.

Figure 1-2. MultiVoice packet format

Ethernet Header	*IP Header*	*UDP Header*	*RTP Header*	*DATA (message)*	*CRC*

For more details on MultiVoice packet processing see Appendix A, "MultiVoice Packet Processing."

Supported audio codecs

MultiVoice provides support for the following audio compression/decompression algorithms (codecs) as defined by the International Telecommunications Union Telecommunications sector standards (Series G) for telephonic audio transmission:

Audio codec	**Description**
G.711	This algorithm transmits and receives a-law and μ–law pulse code modulation (PCM) voice signals at digital bit-rates of 48Kbps, 56Kbps, and 64Kbps. Digital telephone sets on digital PBXs and ISDN channels use this algorithm. Support is required by the H.323 standard. MultiVoice supports both G.711 A-law and G.711 μ–law.
G.723.1	This algorithm performs speech compression/decompression using a low bit rate—5.3Kbps or 6.3Kbps—output quality. This codec is designed specifically for voice transmission over low bit-rate links (greater than 56Kbps). MultiVoice supports this codec at both bit-rates.

Audio codec	Description
G.728	This algorithm performs speech compression/decompression at 16Kbps using low-delay code excited linear predictive methods, with a frame size of 2.5 milliseconds. MultiVoice implements this codec using a frame size of 5 milliseconds. It uses the same bitstream as the ITU-T standard and allows speech processed by a MultiVoice Gateway to be processed by any other gateway that supports the G.728 standard.
G.729(A)	This Conjugate Structure, Algebraic Code Excited Linear Predictive (CS-ACELP) algorithm is used for compression/decompression of speech at 8Kbps, as defined by the ITU-T Standard G.729, with Annex A.
Full-rate GSM	This algorithm is a voice encoder/decoder standard for cellular communications. It compresses the speech samples from 64Kbps PCM to 13.2Kbps, requiring less network than G.711 a-law/μ–law. European, Japanese and Australian cellular communications systems follow this standard, and certain Web phone applications support it. Full-rate GSM uses a speech frame size of 160 samples (20 msec) and the encoder produces 33 bytes per frame. The decoder produces 160 samples (20msec) of speech from the 33-byte encoder output. This algorithm also supports silence detection and comfort noise generation for Full-rate GSM.

H.323 implementation

MultiVoice implements the H.323 standards defined for both gateways and gatekeepers. Gateways connect the PSTN to the IP-based network. Calls originate at a MultiVoice Gateway and travel across the IP network, which are then routed to a second MultiVoice Gateway that is connected to the PSTN and, then ultimately, to the destination phone. The gatekeeper manages the network, supporting all gateways, user profiles, and authentication. The MultiVoice Access Manager (MVAM) performs the gatekeeper functions for a MultiVoice network.

Supporting the H.323 direct-call model for Voice over IP networks, MultiVoice implementation includes

- Integrating PSTN and packet networks to complete calls
- Using primary and secondary gatekeepers
- Using overlapping gateway coverage areas

Integrating PSTN and packet networks

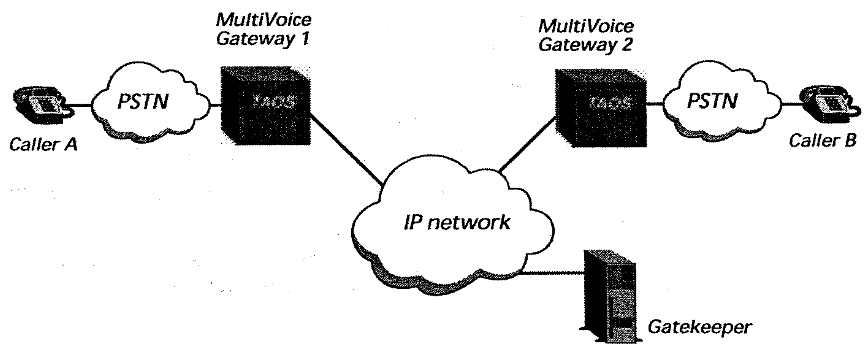
A MultiVoice network integrates both the PSTN and packet networks. Two gateways connect Caller A to Caller B. An NT or Solaris-based server running MVAM is the gatekeeper.

In Figure 1-3, when Caller A dials Caller B, the following high-level events occur:

1. Caller A dials Gateway 1 and enters the PIN authentication (if required) and Caller B's phone number.
2. Gateway 1 establishes a session with the gatekeeper.
3. Gateway 1 forwards the phone number and PIN authentication to the gatekeeper.
4. The gatekeeper authenticates Caller A and, if authentication is successful, forwards the IP address of Gateway 2 to Gateway 1.
5. Gateway 1 establishes a session with Gateway 2.
6. Gateway 2 forwards the call request to Caller B.

When Caller B answers the phone (goes off-hook), voice traffic is transmitted in IP packets between Gateway 1 and Gateway 2 using RTP protocol.

Figure 1-3. Example of a MultiVoice network



If the callers in Figure 1-3 used a traditional voice communications network, Caller A would require a long-distance carrier's services to reach Caller B. But, Caller A is in Gateway 1's coverage area, and can reach the gateway with a local call. The IP-routed network performs the same function as a long-distance carrier's circuit-switched network.

Coverage areas

Each MultiVoice Gateway services a *coverage area*. The coverage area consists of a group of telephone numbers that may dial and receive calls through a particular gateway. Coverage areas for each gateway are defined by assigning dial strings, such as country codes, area codes, country code/area code combinations, area code/exchange combinations, or complete telephone numbers, to a database on the gatekeeper.

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Inclusion areas

Individually, each of the telephone numbers and dial strings assigned to a coverage area represents an individual *inclusion area*. Together, these inclusion areas represent the coverage area for a MultiVoice Gateway. For example, an inclusion area could be specified by the partial telephone number *1732*. This number is composed of a country code of *1* and area code of *732*. A gateway with this inclusion area would cover all telephone numbers within the 732 area code.

Overlapping gateway coverage areas

In a MultiVoice network with overlapping gateway coverage areas, two or more gateways can process incoming calls to telephone numbers in the same coverage area. The MVAM allows you to assign the same inclusion areas, defined by country codes, area codes, country code/area code combinations, area code/exchange combinations, or complete telephone numbers, to two or more gateways, creating overlapping coverage areas.

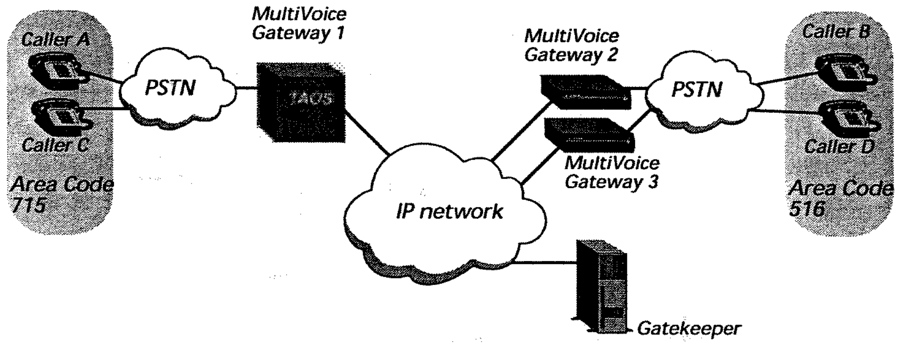
Identical coverage areas may be configured on the gatekeeper for each MultiVoice Gateway in the group. This type of network configuration provides for dynamic call management and allow the gatekeeper to perform call load-leveling across a group of gateways.

Figure 1-4 illustrates a MultiVoice network with overlapping coverage areas. Two gateways provide coverage to area code 516. An NT or Solaris-based server running MVAM is the gatekeeper. When Caller A dials Caller D, Caller C dials Caller B, and both dialed phone numbers are part of the same coverage area, the following high-level events occur:

1. Caller A dials Gateway 1, and enters the PIN authentication (if required) and Caller D's phone number.
2. Gateway 1 establishes a session with the gatekeeper.
3. Gateway 1 forwards the phone number and PIN authentication to the gatekeeper.
4. The gatekeeper attempts to authenticate Caller A and, if successful, identifies all the MultiVoice Gateways that support the coverage area for Caller D's phone number.
5. The gatekeeper forwards the IP address of Gateway 2 to Gateway 1.
6. Gateway 1 establishes a session with Gateway 2.
7. Gateway 2 forwards the call request to Caller D.
8. Now, Caller C dials Gateway 1, and enters his or her PIN authentication (if required) and Caller B's phone number.
9. Gateway 1 establishes a session with the gatekeeper.
10. Gateway 1 forwards the phone number and PIN authentication to the gatekeeper.
11. The gatekeeper attempts to authenticate Caller C and, if successful, identifies the MultiVoice Gateways that support the coverage area for Caller B's phone number.
12. This time the gatekeeper forwards the IP address of Gateway 3 to Gateway 1.
13. Gateway 1 establishes a session with Gateway 3.

14 Gateway 3 forwards the call request to Caller B.

Figure 1-4. Example of a MultiVoice network with overlapping coverage areas



In Figure 1-4, the gatekeeper, having already routed a call from Caller A to Caller D through Gateway 2, determines that the call from Caller C to Caller B should be routed through Gateway 3 instead of Gateway 2 to keep the call volume balanced.

Since MultiVoice uses one port per call, the gatekeeper attempts to assign calls to each gateway based upon port availability, alternating call assignments between covering gateways.

Using primary and secondary gatekeepers

Figure 1-5 shows an example of a MultiVoice network that uses primary and secondary gatekeepers to manage VoIP network operations. The VoIP network configuration in Figure 1-5 provides the MultiVoice network with redundant call-management capability.

Each MultiVoice Gateway may be configured to register with a secondary gatekeeper when it cannot register with the primary gatekeeper. This enables call processing to continue in the event that the primary gatekeeper cannot be reached by a gateway (redundancy).

As illustrated in Figure 1-5, two MultiVoice gateways can connect Caller A to Caller B. Either of the NT or Solaris-based servers running MVAM can be the gatekeeper.

When Caller A dials Caller B, the following high-level events occur:

1 Caller A dials Gateway 1 and enters the PIN authentication (if required) and Caller B's phone number.

2 Gateway 1 attempts to register with it's primary gatekeeper.

  If the registration fails, Gateway 1 attempts to register with its secondary gatekeeper.

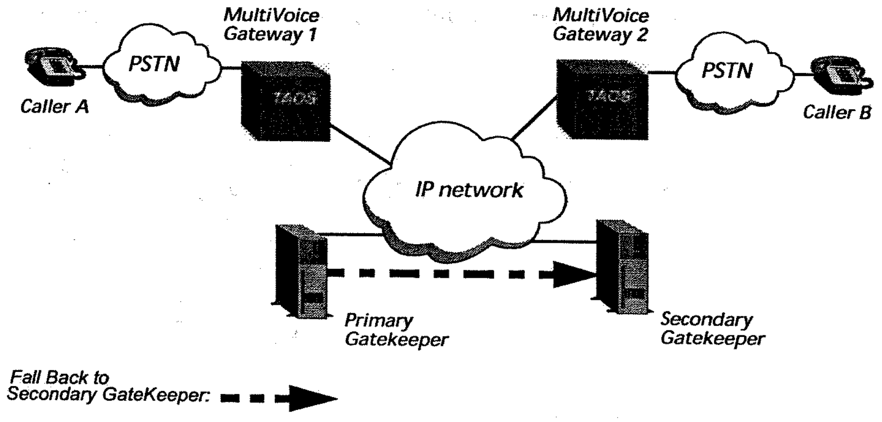
3 When registration is established with the secondary gatekeeper, Gateway 1 forwards the phone number and PIN authentication to the secondary gatekeeper.

4 The secondary gatekeeper authenticates Caller A and, if authentication is successful, forwards the IP address of Gateway 2 to Gateway 1.

5 Gateway 1 establishes a session with Gateway 2.

6 Gateway 2 forwards the call request to Caller B.

Figure 1-5. Example of a MultiVoice network with a secondary gatekeeper



The primary and secondary gatekeepers are separate NT or Solaris-based servers, each with a unique network identity, each running its own copy of the MVAM application, and functioning independently of each other. Each gatekeeper has unique gateway and user databases, and each maintains separate call and activity logs. To ensure coverage, the two gatekeepers must:

- Have duplicate gateway and user information
- Be administered using the same time
- Be synchronized using some third-party clock synchronization mechanism (such as NTP)

The secondary gatekeeper does not report call activity to, nor share call records with the primary gatekeeper. Therefore, if a third-party billing system is used with MultiVoice, all the gatekeepers on the network must communicate with that billing system server.

Increasing gatekeeper reliability

The ITU-T H.323 standard defines a zone as a group of gateways that register with and are administered by a single gatekeeper. Calls may be routed by the gatekeeper directly between any pair of gateways in the same zone.

To maximize gatekeeper reliability, and reliability of a MultiVoice network, multiple MVAM systems, each servicing its own H.323 zone, may be configured as redundant gatekeepers. This is called *sparing*.

In Figure 1-6, MVAM systems in a MultiVoice network are paired for use as reciprocal secondary gatekeepers. Each MVAM serves as both a primary gatekeeper for its selected zone and a secondary gatekeeper for adjoining zones. If a MultiVoice Gateway fails to register with the MVAM in that zone, that gateway attempts to register with its paired MVAM. In this configuration, each gatekeeper maintains duplicate gateway and user information for the reciprocating gatekeeper's zone.

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Figure 1-6. Reciprocal secondary gatekeepers

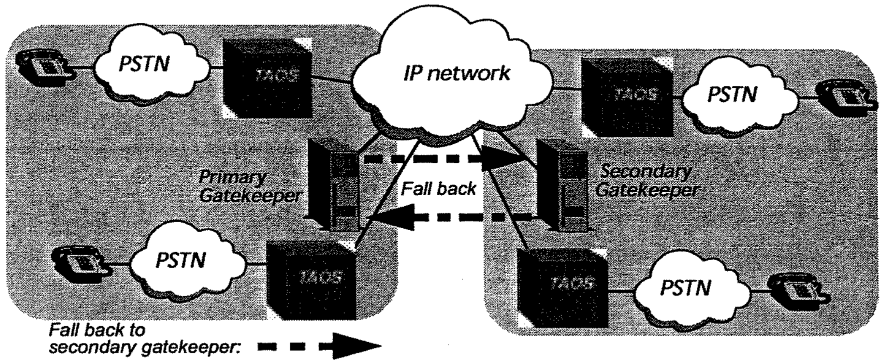
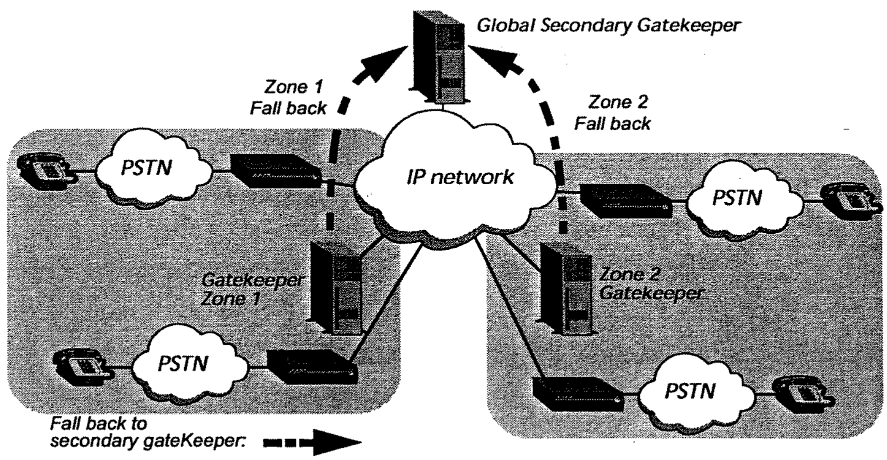


Figure 1-7 illustrates how a MultiVoice network may be configured to use a single MVAM as the secondary gatekeeper for the entire network. If any MultiVoice Gateway fails to register with its primary gatekeeper, that gateway attempts to register with the MVAM performing the global gatekeeper function. The MVAM that performs the global gatekeeper function maintains duplicate gateway and user information for all other gatekeeper zones in the MultiVoice network.

Figure 1-7. Global secondary gatekeeper



Keep-alive registration

Once registered with a gatekeeper, a MultiVoice Gateway must periodically reregister. This is called *keep-alive registration.* Keep-alive registration to informs the gatekeeper that a gateway is available to accept calls. By default, a MultiVoice Gateway attempts keep-alive registration with its primary gatekeeper every 120 seconds. At registration time, the gateway makes up to five registration attempts, at 5-second intervals, until successfully contacting the gatekeeper.

When keep-alive registration fails, the MultiVoice Gateway does one of the following, if:

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